CLAIMS:

1. An adaptive beamformer, comprising:

- a filtered sum beamformer (107) arranged to process input audio signals (u1, u2, u3) from an array of respective microphones (101, 103, 105), and arranged to yield as an output a first audio signal (z) predominantly corresponding to sound from a desired audio source (160), by filtering with a first set of respective adaptable filters (f1(-t), f2(-t), f3(-t)) the input audio signals (u1, u2, u3), the filtered sum beamformer (107) being adaptive in the sense that coefficients of the first set of adaptable filters (f1(-t), f2(-t), f3(-t)) are susceptible to be changed by adding to at least one coefficient a difference value, obtained as a function of an adaptation step size; and
- a scaling factor determining unit (170), arranged to provide a scale factor (S) evaluated as a first function (F1), of a ratio (Q) of a first variable (F2) being an estimate of the non-noise corrupted audio signal originating from the desired sound source (160) present in the first audio signal (z), and a second variable (F3) being an estimate of the noise present in the first audio signal (z),

the adaptive beamformer being arranged to scale the adaptation step size with the scale factor (S).

2. A sidelobe canceller (100) comprising an adaptive beamformer as claimed in claim 1, further comprising:

- an adaptive noise estimator (150), arranged to derive an estimated noise signal (y) by filtering respective noise measurements (x1, x2, x3) derived from the input audio signals (u1, u2, u3) with a second set of adaptable filters (g1, g2); and
- a subtracter (142) connected to subtract the estimated noise signal (y) from the first audio signal (z) to obtain a noise cleaned second audio signal (r).

3. An adaptive beamformer as claimed in claim 1 or a sidelobe canceller as claimed in claim 2, having the coefficients of the first set of filters (f1(-t), f2(-t), f3(-t)) specified in the frequency domain, and being arranged for having the adaptation step size scaled per predetermined frequency range by the ratio (Q) being

$(P_{zz}[f,t] - CP_{A(xi)A(xi)}[f,t]) / P_{zz}[f,t]$,

in which $P_{zz}[f,t]$ is a measure of the power of the first audio signal (z) in the predetermined frequency range around frequency f and for a time instant t, $P_{A(xi)A(xi)}[f,t]$ is a measure of the power of a noise signal derived by a noise estimation unit (310) from at least one noise measurement (x1) by a transformation A, and C is a constant.

4. A sidelobe canceller as claimed in claim 2, having the coefficients of the first set of filters (f1(-t), f2(-t), f3(-t)) specified in the frequency domain, and arranged for having the adaptation step size scaled per predetermined frequency range by the ratio (Q) being $(P_{zz}[f,t] - CP_{A(xi)A(xi)}[f,t]) / P_{rr}[f,t]$,

in which $P_{zz}[f,t]$ is a measure of the power of the first audio signal (z) in the predetermined frequency range around frequency f and for a time instant t, $P_{A(xi)A(xi)}[f,t]$ is a measure of the power of a noise signal derived by a noise estimation unit (310) from at least one noise measurement (x1) by a transformation A, $P_{rr}[f,t]$ is a measure of the power of the second audio signal (r), and C is a constant.

5. An adaptive beamformer as claimed in claim 1, comprising a speech detector (165) providing on the basis of the first audio signal (z) a Boolean designation Speech/Noise, and arranged to adapt the first set of filters (f1(-t), f2(-t), f3(-t)) only if the designation is Speech.

6. A sidelobe canceller as claimed in claim 2, comprising a speech detector (165) providing on the basis of the first audio signal (z) or the second audio signal (r) a Boolean designation Speech/Noise, and arranged to adapt the first set of filters (f1(-t), f2(-t), f3(-t)) only if the designation is Speech.

7. An adaptive beamformer as claimed in claim 1 or a sidelobe canceller as claimed in claim 2, arranged to apply a binary decision function to the ratio (Q), and arranged to adapt the first set of filters (f1(-t), f2(-t), f3(-t)) only if the decision is 1.

8. A handsfree speech communication device comprising an adaptive beamformer as claimed in claim 1 or a sidelobe canceller as claimed in claim 2.

9. A voice control unit comprising an adaptive beamformer as claimed in claim 1 or a sidelobe canceller as claimed in claim 2.

10. A consumer apparatus comprising a voice control unit as claimed in claim 9.

11. A tracking device arranged for tracking an audio producing object, comprising an adaptive beamformer as claimed in claim 1 or a sidelobe canceller as claimed in claim 2.

12. A method of adaptive beamforming, comprising:

- beamforming filtering input audio signals (u1, u2, u3) from an array of respective microphones (101, 103, 105) with a first set of respective adaptable beamforming filters (f1(-t), f2(-t), f3(-t)), yielding a first audio signal (z) predominantly corresponding to sound from a desired audio source (160), the beamforming filtering being adaptive in the sense that coefficients of the first set of adaptable filters (f1(-t), f2(-t), f3(-t)) are changeable by adding to at least one coefficient a difference value obtained as a function of an adaptation step size;
- determining a scale factor (S) a first function (F1), of a ratio (Q) of a first variable (F2) being an estimate of the non-noise corrupted audio signal originating from the desired sound source (160) present in the first audio signal (z), and a second variable (F3) being an estimate of the noise present in the first audio signal (z); and
- scaling the adaptation step size with the scale factor (S).

13. A computer program product comprising respective code for enabling a processor to execute each of the steps of the method of claim 12.